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PATENT APPLICATION

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Docket No: 28944/36228

PATENT APPLICATION TRANSMITTAL UNDER 37 C.F.R. 1.53

Box Patent Application
Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Transmitted herewith for filing is the patent application of

Inventor(s): Pascal SCALART and Franck BOUTEILLE

Title: ADAPTIVE IDENTIFICATION METHOD AND DEVICE, AND ADAPTIVE

1. Type of Application

☒ This is a new application for a

- ☒ utility patent.
- ☐ design patent.

☐ This is a continuation-in-part application of prior application no.

2. Application Papers Enclosed

1	Title Page
14	Pages of Specification (excluding Claims, Abstract, Drawings & Sequence Listing)
5	Page(s) of Claims
1	Page(s) of Abstract
4	Sheet(s) of Drawings (Figs. 1 to 4)

☒ Formal

☐ Informal

Page(s) of Sequence Listing

CERTIFICATION UNDER 37 CFR 1.10

I hereby certify that this Patent Application Transmittal and the documents referred to as enclosed therewith are being deposited with the United States Postal Service on **March 20, 2000**, in an envelope addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231 utilizing the "Express Mail Post Office to Addressee" service of the United States Postal Service under Mailing Label No. EM099900135US.

Richard Zimmermann

RICHARD ZIMMERMANN

3. **Declaration or Oath**

☐ Enclosed

☐ Executed by (check all applicable boxes)

☐ Inventor(s)

☐ Legal representative of inventor(s) (37 CFR 1.42 or 1.43)

☐ Joint inventor or person showing a proprietary interest on behalf of inventor who refused to sign or cannot be reached

☐ The petition required by 37 CFR 1.47 and the statement required by 37 CFR 1.47 are enclosed. See Item 5D below for fee.

☒ Not enclosed - the undersigned attorney or agent is authorized to file this application on behalf of the applicant(s). An executed declaration will follow.

4. **Additional Papers Enclosed**

☐ Preliminary Amendment

☐ Information Disclosure Statement

☐ Declaration of Biological Deposit

☐ Computer readable copy of sequence listing containing nucleotide and/or amino acid sequence

☐ Microfiche computer program

☐ Verified statement(s) claiming small entity status under 37 CFR 1.9 and 1.27

☐ Associate Power of Attorney

☐ Verified translation of a non-English patent application

☐ An assignment of the invention

☒ Return receipt postcard

☐ Other

5. **Priority Applications Under 35 USC 119**

Certified copies of applications from which priority under 35 USC 119 is claimed are listed below and

☒ are attached.

☐ will follow.

COUNTRY	APPLICATION NO.	FILED
France	99 03822	March 26, 1999

6. **Filing Fee Calculation (37 CFR 1.16)**

A. ☒ **Utility Application**

CLAIMS AS FILED - INCLUDING PRELIMINARY AMENDMENT (IF ANY)						
			SMALL ENTITY		OTHER THAN A SMALL ENTITY	
	NO. FILED	NO. EXTRA	RATE	FEE	RATE	FEE
BASIC FEE				$345.00		$690.00
TOTAL	24 -20	= 0	X 9 =	$	X 78 =	$72.00
INDEP.	3 - 3	= 0	X 39 =	$	X 78 =	$
☐ First Presentation of Multiple Dependent Claim			+ 130 =	$	+ 260 =	$
Filing Fee:				$	OR	$762.00

B. ☐ **Design Application ($155.00/$310.00)** Filing Fee: $ ______________

C. ☐ **Plant Application ($240.00/$480.00)** Filing Fee: $ ______________

D. **Other Fees**

☐ Recording Assignment [Fee -- **$40.00** per assignment] $ ______________

☐ Petition fee for filing by other than all the inventors or person on behalf of the inventor where inventor refused to sign or cannot be reached [Fee -- **$130.00]** $ ______________

☐ Other $ ______________

Total Fees Enclosed **$762.00**

7. **Method of Payment of Fees**

☒ Enclosed check in the amount of: $762.00

☐ Charge Deposit Account No. 13-2855 in the amount of: $_____
A copy of this Transmittal is enclosed.

☐ Not enclosed

8. **Deposit Account and Refund Authorization**

The Commissioner is hereby authorized to charge any deficiency in the amount enclosed or any additional fees which may be required during the pendency of this application under 37 CFR 1.16 or 37 CFR 1.17 or under other applicable rules (except payment of issue fees), to Deposit Account No. 13-2855. A copy of this Transmittal is enclosed.

Please refund any overpayment to Marshall, O'Toole, Gerstein, Murray & Borun at the address below.

Please direct all future communications to William E. McCracken, at the address below.

Respectfully submitted,

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March 20, 2000

JOINT INVENTORS **Atty. Docket No. 28944/36228**

"EXPRESS MAIL" mailing label No. EM099900135US.
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I hereby certify that this paper (or fee) is being deposited with the United States Postal Service "EXPRESS MAIL POST OFFICE TO ADDRESSEE" service under 37 CFR §1.10 on the date indicated above and is addressed to: Assistant Commissioner for Patents, Washington, D.C. 20231

Richard Zimmermann

APPLICATION FOR UNITED STATES LETTERS PATENT

SPECIFICATION

TO ALL WHOM IT MAY CONCERN:

Be it known that we, Pascal SCALART, a citizen of France, residing at Les Cinq Croix, 22300, Ploubezre, France, and Franck BOUTEILLE, a citizen of France, residing at 8, rue de Kerniflet, 22300, Lannion, France, have invented a new and useful ADAPTIVE IDENTIFICATION METHOD AND DEVICE, AND ADAPTIVE ECHO CANCELLER IMPLEMENTING SUCH METHOD, of which the following is a specification.

ADAPTIVE IDENTIFICATION METHOD AND DEVICE, AND ADAPTIVE ECHO CANCELLER IMPLEMENTING SUCH METHOD

BACKGROUND OF THE INVENTION

The present invention generally relates to adaptive identification techniques. It is particularly well suited, though not exclusively, to use with echo cancellers used in telecommunications.

Adaptive identification of linear systems characterised by their impulse response has been widely studied and numerous algorithmic solutions have been proposed in specialist literature.

The general problem of direct identification by transverse adaptive filtering, which corresponds to the majority of practical applications, is considered.

Figure 1 illustrates a system to be identified 10, to which a signal x_t varying over time is applied. The response of the system 10 to the input signal x_t is written as z_t. Measurement of the response z_t is inevitably accompanied by the addition of an interference component b_t referred to as observation noise. This observation noise b_t may comprise noise strictly speaking (white noise or traffic noise, for example), but also some useful signal. The component b_t is referred to as observation noise because it interferes with the observation of the response z_t. The adder 12 in the drawing symbolises the superposition of the component b_t, assumed to be additive, to the response z_t. The measured observation signal y_t is therefore the response of a real system 14 comprising the system to be identified 10 and the adder 12.

The adaptive identification device 16 receives the input signal x_t on a first input E1 and the observation signal y_t on a second input E2. The signals x_t and y_t are amplified, filtered and digitised at the input of the device 16 by conventional elements, not illustrated. The adaptive identification device 16 has an identification filter 18 consisting of a programmable filter having a finite impulse response (FIR) expressed as $H_{t-1}^T = \left(h_{t-1}^0, h_{t-1}^1, \cdots, h_{t-1}^{L-1}\right)$, where $(.)^T$ denotes the matrix transposition. The coefficients of the identification filter 18 are adapted so that this impulse response H_{t-1}^T is representative of the impulse response of the

system to be identified 10. The filter 18 receives the digitised input signal x_t and supplies an estimation $\hat{z}_t$ of the response z_t of the system 10.

A subtractor 20 removes this estimation $\hat{z}_t$ from the digitised observation signal y_t to supply an error signal e_t. This error signal e_t may be regarded as an estimation of the interference component b_t.

A updating unit 22 of the identification filter adapts the coefficients of the filter 18 on the basis of the input signal x_t and the error signal e_t, generally taking an adaptation step μ into account.

Numerous algorithms have been proposed to automatically determine the coefficients of the adaptive filter 18. When used in practice, the originators of these devices are generally forced to seek a compromise between the convergence speed of the algorithm, the ease with which it can be controlled and run, the arithmetic complexity and numerical stability.

The LMS algorithm (« *Least Mean Square* ») is the algorithm most widely used to adapt the impulse response of a FIR identification filter continuously over time. Such algorithm provides an efficient implementation of a Wien filter with L coefficients which minimises the mean value of the power of the filtering error in the stochastic approximation. It is defined by the equations:

$$e_t = y_t - X_t^T H_{t-1} \quad (1)$$

$$H_t = H_{t-1} + \mu e_t X_t \quad (2)$$

where $X_t = (x_t, x_{t-1}, \cdots, x_{t-L+1})^T$ represents the vector of the last L samples of the input signal and μ represents the adaptation step of the algorithm. The main advantages of this algorithm are its low numerical complexity, its ease of implementation and its robustness to errors. Unfortunately, if highly correlated signals (such as speech signals) are used to excite the unknown system, this algorithm has a convergence speed which deteriorates rapidly.

In order to get round these problems, a specific version of the LMS algorithm is often used, incorporating a parameter adjustable adaptation step. This algorithm then corresponds to a normalised version of the LMS, or NLMS (« *Normalized Least Mean Square* ») in which the coefficients of the adaptive filter are updated according to the following equation :

$$H_t = H_{t-1} + \frac{\mu e_t}{X_t^T X_t} X_t \qquad (3)$$

Assuming that the optimum filter H^{opt} of the unknown system 10 is a FIR filter of an order lower than or equal to L, equation (3) may be written :

$$\Delta H_t = \left[I - \mu X_t \left(X_t^T X_t \right)^{-1} X_t^T \right] \Delta H_{t-1} - \mu X_t \left(X_t^T X_t \right)^{-1} b_t \qquad (4)$$

where $\Delta H_t = H^{opt} - H_t$ represents the error in the estimated coefficients of the filter at iteration t. This expression corresponds to a geometric interpretation of the NLMS algorithm. In situations where $\mu \neq$, equation (4) corresponds to a relaxed projection of the vector ΔH_{t-1} across an affine sub-space wholly determined by the matrix between the brackets and by knowing the initial shift imparted by the last term of equation (4).

In order to come up with new algorithms offering a higher convergence speed than that of the NLMS algorithm, several approaches have been proposed in the background literature (implementation in the frequency domain, sub-band filtering, ...). Below, we will look at those based on modifying the direction of the projection of the NLMS as well as those based on using a whitening filter.

Improving convergence by modifying the direction of projection

Convergence of the NLMS can be improved by modifying the direction of projection, as mentioned above. This analysis is the basis of affine projection algorithms (APA) which are based on a projection of multiple order equal to P. As a result, the algorithms have much better convergence properties on correlated signals in comparison with the NLMS algorithm (which corresponds to the extreme case where P=1). The P-order APA algorithms (see K. Oseki et al., « An adaptive algorithm using an orthogonal projection to an affine subspace and its properties », Electronics and communications in Japan, 1984, Vol. 67-A, n° 5, pages 19-27) are characterised by updating the coefficients of the identification filter 18 according to the following equations :

$$e_{t,P} = Y_{t,P} - \underline{X}_{t,P} H_{t-1} \qquad (5)$$

$$H_t = H_{t-1} + \mu \underline{X}_{t,P}^{\#} e_{t,P} \qquad (6)$$

where : $\underline{X}_{t,P} = (X_t, X_{t-1}, \cdots, X_{t-P+1})^T$ (7)

$$Y_{t,P} = (y_t, y_{t-1}, \cdots, y_{t-P+1})^T \quad (8)$$

in which $e_{t,P}$ denotes the *a priori* error vector and $\underline{X}^{\#}_{t,P} = \underline{X}^T_{t,P}\left(\underline{X}_{t,P}\underline{X}^T_{t,P}\right)^{-1}$ represents the generalised inverse Moore-Penrose matrix of order L×P. With these equations for updating coefficients of the identification filter, it can be shown that the estimated *a posteriori* error vector $e^{post}_{t,P}$ is equal to :

$$e^{post}_{t,P} = Y_{t,P} - \underline{X}_{t,P} H_t = (1-\mu)e_{t,P} \quad (9)$$

Assuming that the adaptation step of the algorithm is equal to unity, the P order affine projection algorithm cancels out the P *a posteriori* errors defined at equation (9). This latter property explains why the convergence behaviour of the algorithm is very good. Unfortunately, in the basic version described by equations (5) and (6), the theoretical complexity of such algorithms is in the order of $2LP + K_{inv}P^2$ where K_{inv} represents a constant associated with the computation of the inverse matrix incorporated in equation (6), where the parameters L and P denote respectively the number of coefficients of the identification filter and the projection order.

In order to reduce this initial complexity, several fast versions of these algorithms have been proposed which involve segmenting the pseudo-autocorrelation matrix in a manner similar to the fast recursive least squares algorithms. Such techniques enable the initial complexity to be reduced to more reasonable values in the order of 2L + 20P (see : Steven L. Gay, « A fast converging, low complexity adaptive filtering algorithm », Proceedings of the 3rd International Workshop on Acoustic Echo and Noise Control, Plestin-Les-Grèves, France 1993, pages 223-226 ; Steven L. Gay, « Fast projection algorithms with application to voice echo cancellation », Ph.D. Dissertation of the State University of New Jersey, USA, 1994 ; Steven L. Gay et al., « The fast affine projection algorithm », Proceedings of ICASSP'95, pages 3023-3026, 1995 ; M. Montazéri, « Une famille d'algorithmes adaptatifs comprenant les algorithmes NLMS et RLS : application à l'annulation d'écho acoustique », Thèse de Doctorat de l'Université de Paris Sud, 1994 ; M. Tanaka et al., « Reduction of computation for high-order projection algorithm », Electronics

Information Communication Society Autumn Seminar, Tokyo, 1993).

Improving convergence with a whitening filter

Numerous research papers have been devoted to studying ways of improving the performance of adaptive identification systems using predictive structures (see : M. Mboup et al., « LMS coupled adaptive prediction and system identification : a statistical model and transient analysis », IEEE Transactions on signal processing, Vol. 42, n° 10, October 1994, pages 2607-2615 ; S. Benjebara, « Caractéristiques des signaux et capacité de poursuite des non-stationnarités aléatoires : apport des schémas prédictifs et multirésolutions », Thèse de l'Université des Sciences, des Techniques et de Médecine de Tunis II, Tunis, 1997). Case studies have been able to demonstrate two main structures (symmetrical or non-symmetrical) for pre-whitening the excitation signal of the filter using an adaptive filtering technique. The general principle as to how the non-symmetrical structure is processed is illustrated in figure 2.

A structure of this type is essentially based on an empirical approach intended to modify the signal used to update coefficients of the adaptive filter and transform it so as to reduce the conditioning of its auto-correlation matrix (ratio between the maximum and minimum eigenvalues of the auto-correlation matrix of this signal). As a result, coefficients are updated by the adaptation module 22 using the signal available at the output of a linear prediction circuit 24, whereby the prediction is performed from M coefficients. The algorithm applied by the module 22 to update the L coefficients of the identification filter 18 corresponds to the LMS (equation (2)) or to the NLMS (equation (3)). Similarly, the prediction circuit 24 is implemented as an adaptive filter with M coefficients which are updated with the aid of an LMS algorithm.

Studies devoted to analysing the performance obtained from this type of structure have specifically highlighted the existence of a very strong interaction between the prediction and adaptation modules of the system. In particular, this leads to an inter-dependence between the coefficient adaptation equations of the adaptive predictor 24 and those of the identification filter 18. This strong dependence also occurs in the choice of the two adaptation steps μ_P and μ_H, which creates a zone of very low stability of the entire structure.

As a result, the authors mention an instability of the entire predictive

structure identification system, which limits the prediction orders used in the implementations to values which are always less than four so as to ensure a relative stability, thereby limiting the identification performance : it is necessary to select very low adaptation steps μ_P and μ_H which are not compatible with the aim of improving convergence speed.

Reviewing the solutions proposed in the literature to improve the convergence speed of adaptive identification algorithms leads to the conclusion that on the one hand the algorithms based on modifying the projection direction remain complex in terms of the number of arithmetical operations needed to update the filter coefficients if the projection order P is high and that, on the other hand, the predictive structure identification schemes remain very delicate in terms of control and offer a reduced gain in convergence because of the low prediction orders M used in practice to ensure that the global structure remains relatively stable.

An object of the present invention is to propose a method of adaptive identification which has good convergence properties and which is relatively simple to implement, with a limited arithmetic complexity.

SUMMARY OF THE INVENTION

Accordingly, the invention proposes a An adaptive identification method for estimating the response of a system to an input signal, comprising the steps of : receiving the input signal on the one hand and on the other an observation signal, one component of which is said response to the input signal ; determining an error signal e_t at an instant t in accordance with the expression $e_t = y_t - X_t^T H_{t-1}$, where y_t denotes the value of the observation signal at the instant t, H_{t-1} is a column vector made up of L coefficients of a identification filter (18) having a finite impulse response representative of the impulse response of the system, and $X_t^T = \left(x_t, x_{t-1}, \cdots, x_{t-L+1}\right)$ is a row vector made up of the values x_t, x_{t-1},.. , x_{t-L+1} of the input signal at the instant t and at the preceding L-1 instants; and adapting the L coefficients of the identification filter by taking account of the input signal and the error signal. According to the invention, prediction parameters of the input signal are obtained, whereby the energy of a prediction residue on successive frames of the input signal is

minimised, and the L coefficients of the identification filter are adapted by adding to the column vector H_{t-1} a column vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$, where U_t is a column vector made up of the L values of the prediction residue of the input signal at the instant t and at the preceding L-1 instants, and λ is a positive or zero coefficient.

In the preferred modes of applying the method :

- the frames of the input signal have a duration of at least 5 ms ;
- the frames of the input signal have mutual overlaps ;
- the input signal is analysed by a linear prediction of order P-1, preferably being equal to 5 or more, and the column vector U_t is given by $U_t = X_t + \sum_{q=1}^{P-1} a_q X_{t-q}$, in which the terms a_q denote the linear prediction coefficients of said analysis. The order P-1 of the linear prediction may also be a function of an estimated stationarity of the input signal ;
- the input signal is a speech signal reconstituted by a decoder from an incoming binary stream and the prediction parameters of the input signal are extracted from the incoming binary stream by the decoder, which may also supply the prediction residue for adapting the L coefficients of the identification filter.

Another aspect of the invention relates to an adaptive identification device for a system to which an input signal is applied comprising: a first input receiving the input signal; a second input receiving an observation signal, one component of which is a response of the system to the input signal; an identification filter with a finite impulse response of the system; a subtractor producing an error signal e_t expressed by equation (1) above; means for obtaining prediction parameters from the input signal, whereby the energy of a prediction residue on successive frames of the input signal is minimised ; and adaptation means updating the L coefficients of the identification filter by adding to the column vector H_{t-1} a column vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$.

Another aspect of the invention relates to an adaptive echo canceller for removing the echo components of a direct signal from a return signal,

comprising an identification device of the type mentioned above, the first input of which receives the direct signal as the input signal and the second input of which receives the return signal as the observation signal, the error signal constituting the output signal of the echo canceller.

BRIEF DESCRIPTION OF THE DRAWINGS

Figures 1 and 2, discussed above, are block diagrams of adaptive identification devices known from the prior art;

Figure 3 is a block diagram of an echo canceller incorporating an adaptive identification device as proposed by the invention; and

Figure 4 is a block diagram of another embodiment of an echo canceller as proposed by the invention.

DESCRIPTION OF PREFERRED EMBODIMENTS

In the case of stationary signals, the following equation can be established between the linear prediction coefficients $a_1, \ldots, a_{P-1}$ of order P-1 of the input signal x_t and the inverse $R_{t,P}^{-1}$ of the auto-correlation matrix $R_{t,P} = \underline{X}_{t,P} \underline{X}_{t,P}^T$ of the signal (the matrix $\underline{X}_{t,P}$ being defined by equation (7)) :

$$\begin{pmatrix} 1 \\ a_1 \\ \vdots \\ a_{P-1} \end{pmatrix} = R_{t,P}^{-1} \begin{pmatrix} X_t^T U_t \\ 0 \\ \vdots \\ 0 \end{pmatrix} \tag{10}$$

$$\text{where : } U_t = X_t + \sum_{q=1}^{P-1} a_q X_{t-q} \tag{11}$$

Assuming that the APA algorithm or order P defined above has an adaptation step of $\mu=1$, cancellation of the errors *a posteriori* (equation (9)) simplifies the *a priori* error vector (5) : $e_{t,P} = (e_t, 0, \cdots, 0)^T$, so that the adaptation formula (6) of the identification filter becomes, taking account of equation (10) :

$$H_t = H_{t-1} + \frac{e_t}{X_t^T U_t} U_t \tag{12}$$

On this basis, a variant of the APA algorithm of order P can be defined, referred to hereafter as the « pseudo-APA algorithm of order P », wherein an

adaptation step μ is re-introduced and optionally a positive regularisation constant λ. The error signal in the pseudo-APA algorithm is defined by equation (1) and the identification filter is updated according to :

$$H_t = H_{t-1} + \frac{\mu e_t}{X_t^T U_t + \lambda} U_t \qquad (13)$$

The adaptation step μ is typically between 0.1 and 1, and $\lambda \geq 0$. Compared with the prediction-based identification structure described above, the pseudo-APA algorithm of order P has two major differences :

- the prediction coefficients a_q are no longer evaluated at the rate of the samples but at a rate of blocks. They correspond to the minimisation of the linear prediction energy on frames having a size of N samples. Efficient implementation techniques can be used to estimate the set $\{a_q ; q=1, \ldots, P-1\}$ of coefficients of the predictor filter.
- equation (13) used to update coefficients of the identification filter H_{t-1} does not correspond to that used in the studies previously carried out. Indeed, those studies rely on an empirical approach aimed at whitening the excitation signal in order to improve the convergence speed of the modes, this being proportional to the conditioning of the auto-correlation matrix of the excitation signal. As a result, the LMS or NLMS algorithms proposed by the authors as a means of updating coefficients of the identification filter (and also of the predictor) do not correspond to that of equation (13). These algorithms differ due to the normalisation term contained in the equation used to update the filter coefficients, the term expressed as $\left(U_t^T U_t\right)^{-1}$ in the LMS being replaced in equation (13) by a composite term $\left(X_t^T U_t\right)^{-1}$ or $\left(X_t^T U_t + \lambda\right)^{-1}$.

The method proposed by the invention resolves the problem frequently caused by the strong correlation of the signals to be processed. Indeed, in the latter case, the conventional algorithms tend to lose their good convergence properties. The method proposed by the invention enables these good convergence properties to be intrinsically conserved, even in the presence of highly correlated signals, which frequently occur in practice. It also enables a limited arithmetic complexity to be conserved, which is a major advantage for

the real time implementation of these identification algorithms with digital signal processors.

The pseudo-APA algorithm forms the basis of a new family of adaptive identification devices which can be used in various application fields (echo cancellation, propagation channel equalisation, automatic process control, etc.). The case of acoustic echo cancellation will be discussed below by way of non-limiting example.

Figure 3 illustrates an adaptive echo canceller incorporating an adaptive identification device 16 proposed by the invention. The echo canceller is associated with a hands-free telephone system. The input signal x_t, received on the input E1 of the device, is the direct signal destined for the loudspeaker 11 of the hands-free system. The observation signal y_t received on the input E2 of the device is the return signal picked up by the microphone 13 of the hands-free system. This observation signal y_t incorporates echo components z_t from the direct signal and interference components b_t which might include noise and the speech emitted by the speaker. This being the case, the system to be identified consists of the echo path or paths between the loudspeaker 11 and the microphone 13.

The output signal from the echo canceller is the error signal e_t supplied by the subtractor 20 from the observation signal y_t and the echo estimation $\hat{z}_t$ produced by the identification filter 18.

The time windows of the input signal x_t are managed by a module 30 which stores at least the last L samples $x_t, x_{t-1}, \ldots, x_{t-L+1}$. These L samples, which form the vector X_t, are applied : (i) to the identification filter 18 which performs the filtering according to the last term of equation (1) ; (ii) to the adaptation module 22 which updates the coefficients of the filter 18 in accordance with equation (13) ; and (iii) to the prediction filter 32 which applies to the adaptation module 22 the vector U_t defined by equation (11), made up of the last L values of the prediction residue.

At each instant t, the filter 32 merely has to produce the first component of the vector U_t, i.e. the current value of the residue equal to $x_t + \sum_{q=1}^{P-1} a_q x_{t-q}$,

the other components having been calculated and stored on reception of the preceding samples.

In the example illustrated in figure 3, the prediction filter 32 has a conventional trellis structure, the prediction coefficients a_q $(1 \leq q < P)$ being represented by the P-1 associated reflection coefficients $r_1, \ldots, r_{P-1}$. The reflection coefficients r_i are obtained by a module 34, for example, in accordance with the well-known Levinson-Durbin algorithm based on the auto-correlations $\phi(i)$ of the input signal calculated by a module 36. The Levinson-Durbin algorithm run by the module 34 may be expressed as follows :

$E(0) = \phi(0)$

For i ranging from 1 to P-1, do :

$$r_i = \left[\phi(i) + \sum_{j=1}^{i-1} a_j^{i-1}.\phi(i-j) \right] \Big/ E(i-1)$$

$$a_i^i = -r_i$$

$$E(i) = \left(1 - r_i^2\right).E(i-1)$$

For j ranging from 1 to i-1, do :

$$a_j^i = a_j^{i-1} - r_i.a_{i-j}^{i-1}$$

End For

End For

The reflection coefficients r_i applied to the filter 32 are those obtained after iteration P-1, as well as the coefficients a_q of equation (11) $(a_q = a_q^{P-}\ )$ The quantity E(P-1) is the energy of the residual prediction error.

The correlation coefficients $\phi(i)$ are calculated by the module 36 as follows :

$$\phi(i) = \sum_{j=0}^{N-1} \tilde{x}(j).\tilde{x}(j-i) \qquad (14)$$

where $\tilde{x}(j) = x_{t-j}.f(j)$ represents the input signal multiplied by a conventional windowing function such as a rectangular, Hamming or other function.

The calculation of the correlations $\phi(i)$ and the Levinson-Durbin

algorithm, which is equivalent to a minimisation of the prediction energy E(P-1), are executed on frames of N samples of the input signal, N being a number of the same order of magnitude as the length L of the impulse response of the identification filter 18.

By way of example, if the signals are sampled at a frequency Fe = 8 kHz, the filter 18 may have L = 256 coefficients and the size of the frames may be N = 160, i.e. the frames are 20 ms long. In a telephony application, it can be assumed that over such a duration the speech signals are almost stationary, which validates one of the assumptions made in deriving the pseudo-APA algorithm from the APA algorithm. In general, this frame duration may be longer than 5 ms.

The frames of the input signal processed by the predictive analysis modules 34, 36 preferably overlap, which will enable the non-stationary aspect of the signal to be taken into account. The predictive analysis is then performed every K samples of the input signal where K < N. By way of example, the duration of overlap between two successive frames may be in the order of 15 ms. In the case of the numerical example above, this corresponds to K = 40, each frame being made up of four successive blocks.

If N is a multiple of K, this will further simplify calculation of the correlation coefficients by module 36. After receiving each block of K values of the input signal x_t, it will be sufficient to calculate a partial correlation for each i corresponding to the K most recent terms of equation (14) and to update the correlations $\phi(i)$ by adding to them the partial correlations just calculated and removing from them the partial correlations calculated N/K blocks before.

It should be pointed out that other techniques may be used to implement the prediction filter. For example, a method other than the Levinson-Durbin algorithm may be used to calculate the reflection coefficients r_i or a different structure for the prediction filter based on the reflection coefficients r_i. It is also possible to use another known structure of the prediction filter based directly on the prediction coefficients a_q, or on the LAR coefficients (« Log-Area-Ratio », $LAR_i = \log_{10}[(1-r_i)/(1+r_i)]$), or alternatively on the LSP coefficients (« Line Spectrum Pair »), ...

The device proposed by the invention offers the possibility of varying the order P-1 of the prediction as a function of the instantaneous characteristics

of the input signal. In particular, the stationarity of the signal x_t may be evaluated, for example by analysing the correlation or partial correlation coefficients calculated by the module 36 in order to adopt a relatively high prediction order P-1 if the input signal tends towards being stationary and a lower prediction order in the presence of non-stationarities.

It is possible to evaluate the arithmetic complexity of the pseudo-APA algorithm. The relevant characteristics are set out in Table I, where they are compared with those of the exact NLMS and APA algorithms of order P, in a configuration which is typical of echo cancellation. In the table, P denotes the maximum order of the pseudo-APA algorithm which the circuit 16 can implement and P' the order effectively used in the comparative example.

	NLMS	exact APA	Pseudo-APA
Theoretical complexity per sample (number of elementary operations)	3L	$2\,LP + K_{inv}\,P^2$	Max{3L, $\frac{N}{2}P + (P'+5).P'$}
Complexity/sample with the configuration : Fe = 8kHz L = 256 coefficients N = 160 (20ms) P' = P-1 = 8	768 op. 6,14 Mflops	> 4096 op. > 32,7 Mflops	824 op. 6,6 Mflops

Table I

It is noted that the arithmetic complexity of the pseudo-APA algorithm is essentially equivalent to that of the NLMS in the case of an identification filter having L = 256 coefficients and for a prediction filter whose coefficients are evaluated every 5 ms on 20 ms frames (K = 40, N = 160). Accordingly, the invention proposes algorithms approaching the performance of high-order affine projection algorithms (typically higher than 4) for a complexity lower than that of the exact affine projection algorithm of order 2.

In terms of hardware, the identification device 16 may be set up from commercial circuits, in particular from real time digital signal processors (DSP) with floating point arithmetic commonly used (e.g. TMS320C3X and

TMS320C4X sold by the company Texas Instruments, AD21061 sold by the company Analog Devices ...). Fixed point arithmetic processors may also be used, the usual precautions being taken with regard to the appropriate framing of the data in the processing.

In the alternative embodiment diagrammatically shown in figure 4, the input signal x_t is a speech signal reconstructed by a decoder 40 from an incoming binary stream Φ. The binary stream Φ was produced by a coder dual to the decoder 40, disposed in the communication network or in the terminal used by the remote speaker.

By way of example, the coder/decoder may correspond to the RPE-LTP codec used in the GSM cellular radio telephony system (see Specification GSM 06.01 issued by the European Telecommunications Standard Institute). This coder, like most digital audio coders currently used, operates on the basis of linear prediction on frames of the speech signal corresponding to an almost stationary zone of the signal, which will fulfil the requirements of this invention. In the case of GSM, the frames are 20 ms. In the binary stream Φ which it issues, the coder incorporates on the one hand quantification parameters for the LAR coefficients mentioned above, which characterise the 10-order linear prediction filter of the speech signal, and, on the other, quantification parameters of an excitation signal corresponding to the linear prediction residue.

The decoder 40 has circuits which recover the excitation signal from its quantification parameters read in the stream Φ and a synthesis filter which is the inverse of the 10-order prediction filter to which the coefficients derived by the quantified LAR's are applied. The synthesis filter receives the excitation signal and issues the decoded speech signal x_t. The decoder 40 can therefore obtain the linear prediction parameters from the stream Φ instead of modules 34 and 36 of the device illustrated in figure 3. It may also provide the prediction residue (corresponding to the excitation signal of the synthesis filter) forming the components of the vector U_t needed for the adaptation module 22.

In an equipment incorporating a speech decoder and an adaptive echo canceller, it may be seen that using the pseudo-APA algorithm as illustrated in figure 4 does not introduce any additional complexity as compared with the NLMS, regardless of the order P.

WE CLAIM:

1. An adaptive identification method for estimating a response of a system to an input signal, comprising the steps of :

- receiving the input signal and an observation signal, whereby the observation signal has a component consisting of said response to the input signal ;
- determining an error signal e_t at an instant t according to: $e_t = y_t - X_t^T H_{t-1}$, where y_t denotes a value of the observation signal at the instant t, H_{t-1} is a column vector made up of L coefficients of a identification filter having a finite impulse response representative of an impulse response of the system, and $X_t^T = (x_t, x_{t-1}, \cdots, x_{t-L+1})$ is a row vector made up of values x_t, $x_{t-1}, \ldots, x_{t-L+1}$ of the input signal at the instant t and at L-1 preceding instants;
- obtaining prediction parameters of the input signal, whereby an energy of a prediction residue on successive frames of the input signal is minimised; and
- adapting the L coefficients of the identification filter by adding to the column vector H_{t-1} a column vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$, where U_t is a column vector made up of L values of the prediction residue of the input signal at the instant t and at the L-1 preceding instants, and λ is a positive or zero coefficient.

2. A method as claimed in claim 1, wherein said frames of the input signal have a duration of at least 5 milliseconds.

3. A method as claimed in claim 1, wherein said frames of the input signal have mutual overlaps.

4. A method as claimed in claim 1, comprising the step of analysing the input signal by a linear prediction of order P-1, and wherein the column vector

U_t is given by $U_t = X_t + \sum_{q=1}^{P-1} a_q X_{t-q}$, where the terms a_q denote linear prediction coefficients resulting from said analysis.

5. A method as claimed in claim 4, wherein the order P-1 of the linear prediction is at least 5.

6. A method as claimed in claim 4, wherein the order P-1 of the linear prediction is a function of an estimated stationarity of the input signal.

7. A method as claimed in claim 1, wherein the input signal is a speech signal reconstituted by a decoder from an incoming binary stream and wherein the prediction parameters of the input signal are extracted from the incoming binary stream by the decoder.

8. A method as claimed in claim 7, wherein the decoder supplies the prediction residue used to adapt the L coefficients of the identification filter.

9. An adaptive identification device for a system to which an input signal is applied, comprising :

- a first input for receiving the input signal ;
- a second input for receiving an observation signal having a component consisting of a response of the system to the input signal ;
- an identification filter having a finite impulse response representative of an impulse response of the system ;
- a subtractor producing an error signal by subtracting from the observation signal the input signal filtered by the identification filter, wherein the error signal e_t produced by the subtractor at an instant t is expressed as $e_t = y_t - X_t^T H_{t-1}$, where y_t denotes a value of the observation signal at the instant t, H_{t-1} is a column vector made up of L coefficients of the identification filter, and $X_t^T = (x_t, x_{t-1}, \cdots, x_{t-L+1})$ is a row vector made up of the values x_t, $x_{t-1}, \ldots, x_{t-L+1}$ of the input signal at the instant t and at L-1 preceding instants ;

- means for obtaining prediction parameters from the input signal, whereby the energy of a prediction residue on successive frames of the input signal is minimised;
- means for adapting the coefficients of the identification filter by adding to the column vector H_{t-1} a column vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$, where U_t is a column vector made up of the L values of the prediction residue of the input signal at the instant t and at the L-1 preceding instants, and λ is a positive or zero coefficient.

10. A device as claimed in claim 9, wherein said frames of the input signal have a duration of at least 5 milliseconds.

11. A device as claimed in claim 9, wherein said frames of the input signal have mutual overlaps.

12. A device as claimed in claim 9, wherein the means for obtaining the prediction parameters of the input signal comprise means for analysing the input signal by a linear prediction of order P-1, and wherein the column vector U_t is given by $U_t = X_t + \sum_{q=1}^{P-1} a_q X_{t-q}$, where the terms a_q are linear prediction coefficients produced by the analysis means.

13. A device as claimed in claim 12, wherein the order P-1 of the linear prediction is at least 5.

14. A device as claimed in claim 12, wherein the order P-1 of the linear prediction is a function of an estimated stationarity of the input signal.

15. A device as claimed in claim 9, wherein the input signal is a speech signal reconstituted by a decoder from an incoming binary stream, and wherein the prediction parameters of the input signal are extracted from the incoming binary stream by the decoder.

16. A device as claimed in claim 15, wherein the decoder provides the prediction residue to the means for adapting the coefficients of the identification filter.

17. An adaptive echo canceller for removing echo components of a direct signal from a return signal, comprising :

- a first input for receiving the direct signal ;
- a second input for receiving the return signal ;
- an identification filter having a finite impulse response representative of an echo component generation in the return signal ;
- a subtractor producing an error signal as an output signal of the echo canceller, by subtracting from the return signal the direct signal filtered by the identification filter, wherein the error signal e_t produced by the subtractor at an instant t is expressed as $e_t = y_t - X_t^T H_{t-1}$, where y_t denotes a value of the return signal at the instant t, H_{t-1} is a column vector made up of L coefficients of the identification filter, and $X_t^T = (x_t, x_{t-1}, \cdots, x_{t-L+1})$ is a row vector made up of the values x_t, $x_{t-1}, \dots,$ x_{t-L+1} of the direct signal at the instant t and at L-1 preceding instants ;
- means for obtaining prediction parameters from the direct signal, whereby the energy of a prediction residue on successive frames of the direct signal is minimised;
- means for adapting the coefficients of the identification filter by adding to the column vector H_{t-1} a column vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$, where U_t is a column vector made up of the L values of the prediction residue of the direct signal at the instant t and at the L-1 preceding instants, and λ is a positive or zero coefficient.

18. An adaptive echo canceller as claimed in claim 17, wherein said frames of the direct signal have a duration of at least 5 milliseconds

19. An adaptive echo canceller as claimed in claim 17, wherein said frames of the direct signal have mutual overlaps.

20. An adaptive echo canceller as claimed in claim 17, wherein the means for obtaining the prediction parameters of the direct signal comprise means for analysing the direct signal by a linear prediction of order P-1, and wherein the column vector U_t is given by $U_t = X_t + \sum_{q=1}^{P-1} a_q X_{t-q}$, where the terms a_q are linear prediction coefficients produced by the analysis means.

21. An adaptive echo canceller as claimed in claim 20, wherein the order P-1 of the linear prediction is at least 5.

22. An adaptive echo canceller as claimed in claim 20, wherein the order P-1 of the linear prediction is a function of an estimated stationarity of the direct signal.

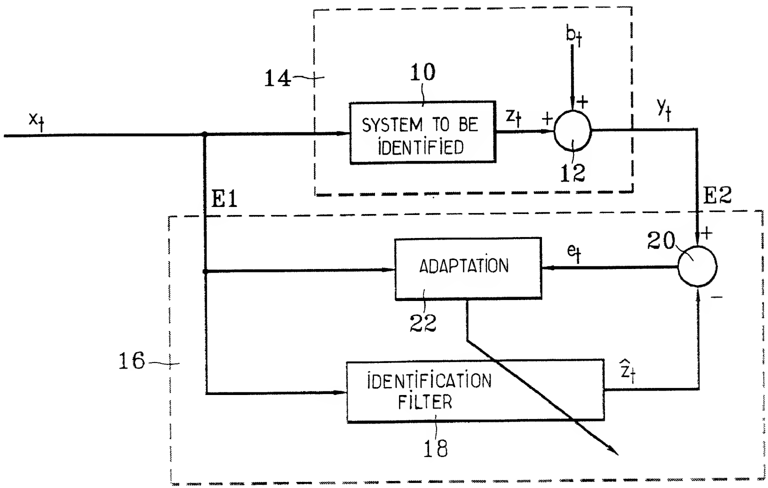
23. An adaptive echo canceller as claimed in claim 17, wherein the direct signal is a speech signal reconstituted by a decoder from an incoming binary stream, and wherein the prediction parameters of the direct signal are extracted from the incoming binary stream by the decoder.

24. An adaptive echo canceller as claimed in claim 23, wherein the decoder provides the prediction residue to the means for adapting the coefficients of the identification filter.

ABSTRACT OF THE DISCLOSURE

The device receives an input signal x_t and an observation signal y_t, one component of which is the response of a system to the input signal. It determines an error signal e_t at an instant t in accordance with the equation $e_t = y_t - X_t^T H_{t-1}$, where H_{t-1} is a vector made up of L coefficients of an identification filter having a finite impulse response representative of the impulse response of the system and $X_t^T = (x_t, x_{t-1}, \cdots, x_{t-L+1})$. A predictive analysis of the input signal is performed by a computation applied to frames of the input signal. The L coefficients of the identification filter are adapted by adding to the vector H_{t-1} a vector proportional to $\frac{e_t}{X_t^T U_t + \lambda} U_t$, where U_t is a vector made up of the L values of a prediction residue of the input signal at the instant t and at the preceding L-1 instants, and λ is a positive or zero coefficient.

FIG. 1



14
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x_t
z_t
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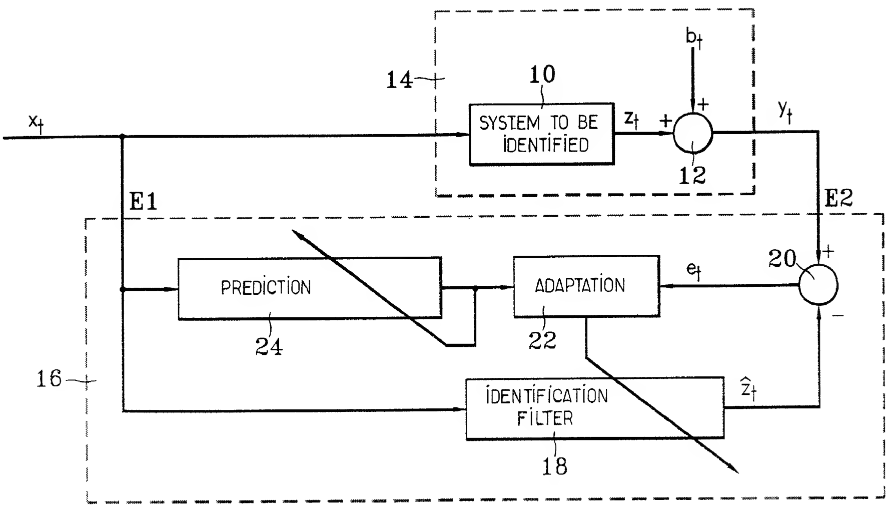
y_t
E1
E2
20
e_t
ADAPTATION
22
16
IDENTIFICATION FILTER
18
$\hat{z}_t$

FIG. 2



14
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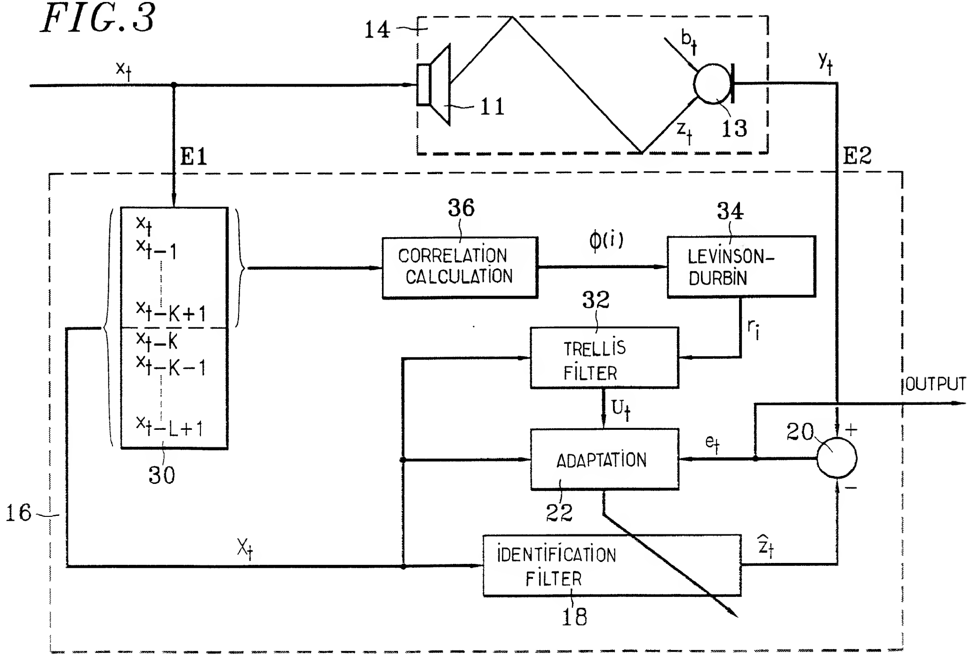
z_t
b_t
+
+
12
y_t
E1
E2
PREDICTION
24
ADAPTATION
22
e_t
20
+
−
16
IDENTIFICATION FILTER
18
$\hat{z}_t$

FIG.3



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b_t
y_t
z_t
13
E1
E2
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x_{t-K+1}
x_{t-K}
x_{t-K-1}
x_{t-L+1}
30
36
CORRELATION CALCULATION
$\phi(i)$
34
LEVINSON-DURBIN
32
r_i
TRELLIS FILTER
U_t
OUTPUT
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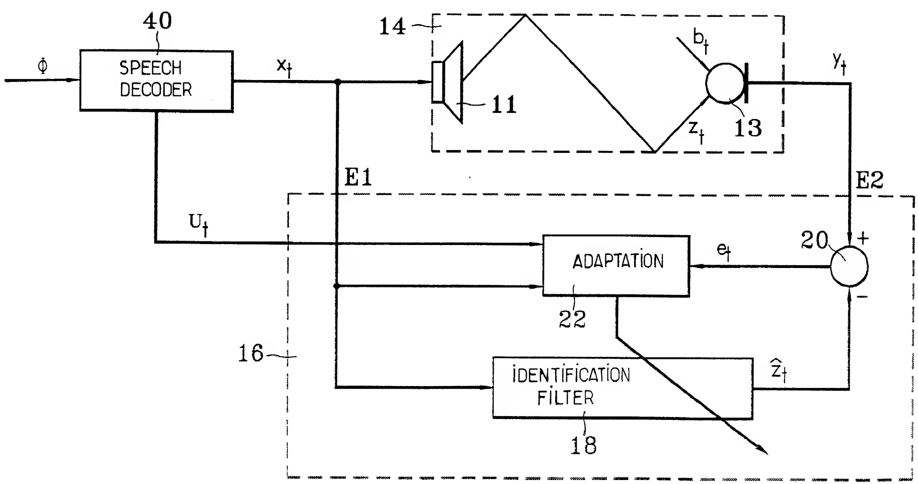
+
e_t
ADAPTATION
-
22
16
X_t
IDENTIFICATION FILTER
$\hat{z}_t$
18

FIG. 4



Φ
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U_t
14
11
b_t
z_t
13
y_t
E1
E2
16
ADAPTATION
22
e_t
20
+
-
IDENTIFICATION FILTER
18
$\hat{z}_t$